Quick Setup Guide

Aastra MX-ONE™ 5.0 Integration with Microsoft® Exchange Server 2013 UM

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Date: January 2014
Doc no.: ASE/MXO/PLM/0366/0/1/EN Rev A

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1 Introduction

The Aastra MX-ONE communication system is based on an open software and hardware environment, using standard servers with a Linux SUSE operating system. This open standards approach enables Aastra to offer our customers a choice and with this in mind we have worked together with Microsoft to ensure that Aastra MX-ONE can be integrated with the latest Microsoft UC products.

1.1 General

MX-ONE 5.0 can interwork with third party UC products using standards-based protocols, such as SIP and CSTA V3/XML.

Integration of MX-ONE 5.0 SP4 with the Microsoft Exchange Server 2013 Unified Messaging (UM) as a complementary solution providing end user services like voice mail, Unified Messaging and auto attendant as well as system functionalities such as load balancing and fault tolerance.

Microsoft Partner Program has certified the integration between MX-ONE 5.0 SP4 and Microsoft Exchange Server 2013 Unified Messaging (UM) via a Direct SIP connection.

1.2 Scope of this document

The intent of this guide is to describe the basic integration between the Aastra MX-ONE and Microsoft Exchange Server 2013 Unified Messaging as well as describe the configuration needed and what features are available after the integration. The following sections describe the solution integration that has been certified through the Microsoft partner program and also the tests performed in Aastra´s laboratory.

For a more technical description on how this integration is set-up, as well as tested features, we refer to the relevant CPI documentation for MX-ONE or please, go to the Microsoft Exchange Server 2013 product websites.

Please, always check the latest products documentation.

2 Solution Description

The integration of MX-ONE 5.0 and Microsoft Exchange Server 2013 Unified Messaging described in this guide is achieved via Direct SIP.
Direct SIP that is specified by Microsoft means that a SIP trunk is used to connect MX-ONE Telephony System 5.0 SP4 or later and Microsoft Exchange Server 2013 Unified Messaging. Additionally, MX-ONE can be configured with TLS and SRTP when integrated with Exchange 2013 UM to provide security in the transport between the systems as well as load balancing and failover functionalities.

2.1 MX-ONE 5.0 SP4 integration with Microsoft Exchange Server 2013 UM

The solution diagram below shows how MX-ONE is connected with Exchange 2013 UM.
In the validated scenario both Client Access and Mailbox role run in the same Exchange Server.

Please note that the Microsoft Exchange Server 2013 architecture is different than the architecture in Exchange Server 2010, read Microsoft document “Voice Architecture Changes for more information.
http://technet.microsoft.com/en-us/library/jj150516(v=exchg.150).aspx

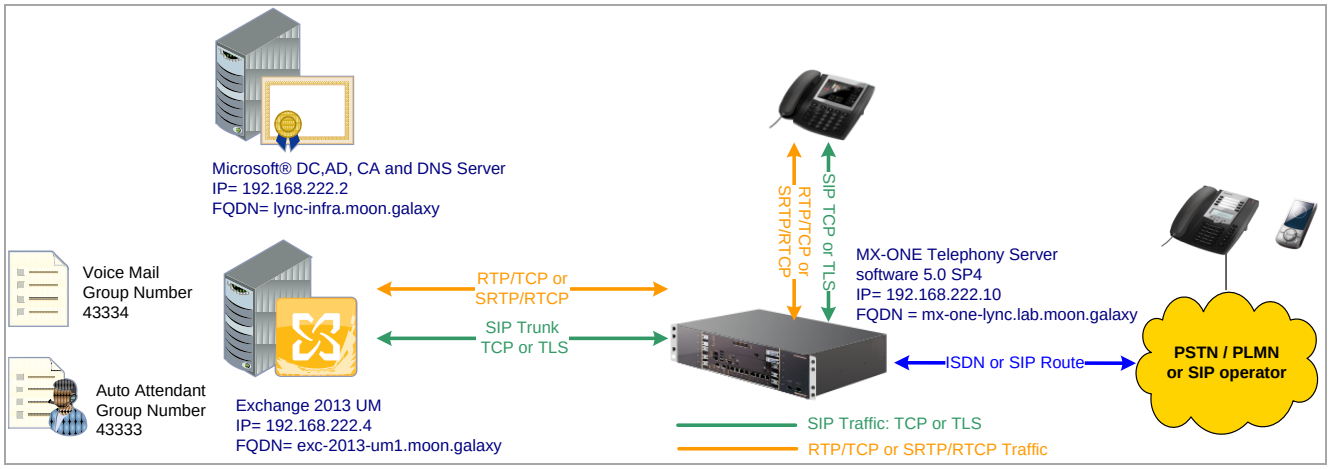


Figure 1 - MX-ONE Telephony System 5.0 SP4 integration with Microsoft Exchange Server 2013 UM

As described in Microsoft´s documentation: "*In the new model, the Client Access server running the Microsoft Exchange Unified Messaging Call Router service redirects Session Initialization Protocol (SIP) traffic that's generated from an incoming call to a Mailbox server. Then a media (Realtime Transport Protocol (RTP) or secure RTP (SRTP)) channel is established from the VoIP gateway or IP Private Branch eXchange (PBX) to the Mailbox server that hosts the user's mailbox.*"

In short, the Direct SIP integration works in the following way: When MX-ONE Telephony System is configured to use TCP as transport, it calls to Microsoft Exchange 2013 UM by sending a SIP INVITE message to the 5060 port of Exchange Server. Then, Exchange Server sends 302 (Moved Temporarily) back to MX-ONE asking to send the INVITE on a different port (TCP: for example, 5065 or 5067). After the MX-ONE sends the INVITE to the new port, the call setup is executed and the call is established.

MX-ONE Telephony System integrated with Microsoft Exchange Server 2013 Unified Messaging delivers the following end user features:

- Voice mail
- Auto Attendant
- Message waiting indication for MX-ONE terminals
- Outlook voice access

3 Licenses

3.1 MX-ONE

The Aastra MX-ONE licenses needed for this integration are:

- SIP trunk licenses. At a minimum there is the need for one SIP trunk license (SIP route) per Microsoft Exchange 2013 UM server. Please, note that the actual quantity of licenses will depend on the customer installation.
- The Automatic-Registration license, which enables the remote SIP extension capability, is mandatory for this integration. This is a system wide license regardless of the number SIP trunks.
- IP extension licenses. The number of licenses depends on the number of pilot numbers created towards the Microsoft Exchange 2013 UM server. The minimum pilot numbers needed for this integration is two.
- An optional VoIP Encryption license is required if security (TLS/SRTP) is used.

Please, always check with your Aastra partner that your system has the correct licenses, before beginning the integration deployment.

3.2 Microsoft Exchange 2013 UM

Microsoft licenses needed for this integration are not included as part of the scope of this guide. Please, contact Microsoft or a qualified Microsoft partner to obtain the proper license requirements for each component of the Microsoft Exchange 2013 UM solution.

4 MX-ONE Integration with Exchange 2013 UM using TCP

4.1 Prerequisites

4.1.1 MX-ONE Prerequisites

Main components

Aastra MX-ONE Telephony System 5.0 SP4 or later with the proper licenses.
At least the following MX-ONE components are required:

- Aastra MX-ONE communications system
  - Telephony Server
    - MX-ONE Telephony Server 5.0 SP4 or later
  - Supported media gateways with the latest compatible firmware with MX-ONE 5.0 SP4 or later
    - MX-ONE Classic - 7U 19-inch chassis, using MGU boards or
    - MX-ONE Lite - 3U 19-inch chassis, using MGU board
    - MX-ONE Slim - 1U 19-inch chassis, using MGU board

The following shall be configured:

- Trunk between MX-ONE and Exchange UM - SIP route.
- Two IP extensions numbers to be used as Pilot numbers (groups) in Exchange UM.
- Message Waiting Indicator configuration in the system and in the phones that will use the service.
- Call list for IP phones. This feature is used to forward the call to the voice mail in case of no answer or busy.

The following MX-ONE type of devices can be used with Exchange 2013 UM:

- SIP – Aastra 67xxi family or any device supporting baseline SIP. As the Exchange Server also supports SIP with Direct Media, MX-ONE gateway resources would not be needed for SIP devices. But, in order to guarantee interoperability with any 3rd party SIP terminal, the SIP route to Exchange UM can be setup as “forced gateway”. The effect is that SIP calls to the Exchange UM server will always transit via the MX-ONE media gateway (MGU) for a call setup and media.
- Non SIP – All non SIP devices calling into the Exchange UM server will transit via the MX-ONE Media GW (MGU based) for call setup and media. The following is the list of supported devices:
  - H.323 - Aastra Dialog 4400 IP phones and Aastra 7400 IP phones (incl. Dialog 5446 Premium)
  - Digital phones: Aastra Dialog 4200 series digital phones
  - Analog phones: Aastra Dialog 4100 series analog phones
  - Aastra Cordless Phones: DT690, DT390, DT412, DT422, DT432
  - Mobiles devices (no MWI functionality) using MX-ONE’s Mobile extension service
  - External callers coming in via the MX-ONE public access, regardless of the type of terminal or network connection (SIP or TDM)

4.1.2 Microsoft Exchange 2013 UM Prerequisites

This guide does not cover the Exchange 2013 UM installation, so our recommendation is that Microsoft Exchange 2013 UM shall be installed by a trained Microsoft engineer.

Before you start to install Microsoft Exchange 2013 Unified Messaging, please read the Microsoft Exchange 2013 documentation for a better understanding of the solution requirements.
The documentation can be found in the following links:

- Microsoft Exchange 2013 documentation
  http://technet.microsoft.com/en-us/library/bb124558(v=exchg.150).aspx
- Microsoft Exchange 2013 Unified Messaging
  http://technet.microsoft.com/en-us/library/jj150478(v=exchg.150).aspx

4.2 Configuration

In this configuration example, we have used the following information:

Direct SIP connection using TCP as transport

MX-ONE Telephony System

- IP address: 192.168.222.10
- FQDN: mx-one-lync.lab.moon.galaxy
- Numbering Plan: 5 digits
- IP extensions numbers for Voice Mail and Auto Attendant: 43334 and 43333.
- Route access code: 043
- Users IP extensions: 27000, 27001 and 27010.

Exchange UM

- IPv4: 192.168.222.4
- FQDN: exc-2013-um1.moon.galaxy
- Voice Mail Pilot identifier, Hunt Group: 43334
- Auto Attendant: 43333

4.2.1 MX-ONE configuration

Voice Mail and Auto Attendant Numbers

The Voice Mail and the Auto Attendant numbers need to be initiated. In this example, the service number 43334 is used for Voice Mail and service number 43333 is used for Auto Attendant.

Number Initiation:
number_initiate -numbertype EX -number 43333
number_initiate -numbertype EX -number 43334

Creating SIP trunk

The following commands shall be executed in MX-ONE to configure a SIP Trunk.

Basic Setup:
rocai:rou=55,sel=7110000000000010,sig=0111110000A0,traf=03151515,trm=4,serv=3100000001,bcap=000100; #varc d5=1 (forced gateway) #varo d6=9 (9s timeout for 100 Trying)
rodai:rou=55,type=TL66,vari=00000011,varc=00001000,varo=00000900;

Please note that Message Waiting Indication number needs to be defined in the SIP route via mwinumber parameter as shown in the example.

Outbound Setting:
sip_route -set -route 55 -uristring0 sip:?@192.168.222.4 -remoteport 5060 -fromuri0 sip:?@192.168.222.10 -protocol tcp -codecs PCMA,PCMU -mwinumber 43334 # mwinumber is the Message Waiting Indication number

Inbound Setting:
sip_route -set -route 55 -accept REMOTE_IP -match 192.168.222.4 Note that –accept REMOTE_IP will match the IP address send in the IPv4 source IP header.

Roeqi:
#Node 1 (as in node x, as in TRU=x-1) for MX-ONE SIP access, in this case the IP address 192.168.222.10 is configured in the Telephony Server 1. roeqi:rou=55,tru=1-1;
Define external destination SIP route data RODDI:ROU=55,DEST=043,ADC=000500000000025000000101000O,SRT=2;

Creating IP extensions and extension category

As Voice Mail in MX-ONE is a special purpose extension, the service numbers are configured as extensions in MX-ONE. We choose “remote extension over sip route” as SIP route allows to filter codecs and have “forced gateway” for improved interoperability (which is not dependent on certain SIP terminal capability).

Extension Profile:
extension_profile -i --csp 2 --ext-serv 2101021000013000000 --ext-traf 0103151515 --ext-cdiv 111000001110 --ext-roc 222201--ext-npres 001100
extension -i -d 43333 --csp 2 -l 1
extension -i -d 43334 --csp 2 -l 1

When the following extensions are called from MX-ONE users, the call will be setup on route 55 that is the route to Exchange Server.

Voice Mail extension 43334:
ip_extension -i -d 43334 --terminal-identity "sip:43334@192.168.222.10" --uri "ROU:55;remote-number=43334"

Auto attendant extension 43333:
ip_extension -i -d 43333 --terminal-identity "sip: 43333@192.168.222.10" --uri "ROU:55;remote-number=43333"

User configuration to forward to Voice Mail

Any third party terminal registered in MX-ONE may subscribe on Message Waiting Indicator (MWI) according to RFC 3842.
The commands below enable a user to forward calls to Exchange Server voice mail.
The example shows how calls will be forwarded to Exchange 2013 UM Voice Mail number 43334 if a call is made to extension 27000 on no answer.

Call List Setup:
call_list -i -d 27000 --dest-number 27000 --position 1 --busy-position 2
call_list -i -d 27000 --dest-number 43334 --position 2 -- ird –bypass true

For non generic extensions it is recommended to use Call Diversion.

Enable Voice Mail and Voice Mails notification in the MX-ONE Telephony System (MWI settings)

The voice mail extension 43334 will be used as the handle (attribute mwf), where to feed incoming Message Waiting Indicator (MWI), as defined in RFC 3842. The message from Exchange 2013 UM

includes the SIP account from the user's extension that needs to receive the message waiting indication; that there is a voice mail to listen to.
Create the attribute mwf (message waiting function) and attach it to the voice mail extension. In ICMWC set the number to be called from an H.323 terminal in the dig parameter.

Message Waiting Settings:
Icfui:ifcind=2,user=generic,istype=2,inttyp=1;
icfuc:mwf=all;
icmwc:sid=02,dtxt=43334,dig=43334,kfcn=MWC;
vmgei:ifcind=2,dir=43334;

*For H.323 Dialog terminals: If there is no fixed key for Voice Mail on the terminal, a function key, 'Message Waiting' must be enabled in order to enable speed dial. The key is enabled in a common phone configuration file (for example d42x02-config.txt).
For SIP 67xxi terminals: In the common phone configuration file, aastra.cfg, set "sip line1 vmail: 43334" to enable speed dial to voice mail.

4.2.2 Microsoft Exchange 2013 UM

In order to setup the Exchange 2013 UM, please check Microsoft´s documentation:
Deploy Exchange 2013 UM
http://technet.microsoft.com/en-us/library/jj673564(v=exchg.150).aspx

After the installation of the Exchange 2013 UM roles, the following steps need to be executed to create the integration between MX-ONE and Exchange 2013 UM.

UM Dial Plan

A "UM Dial Plan" needs to be created in the Exchange UM.

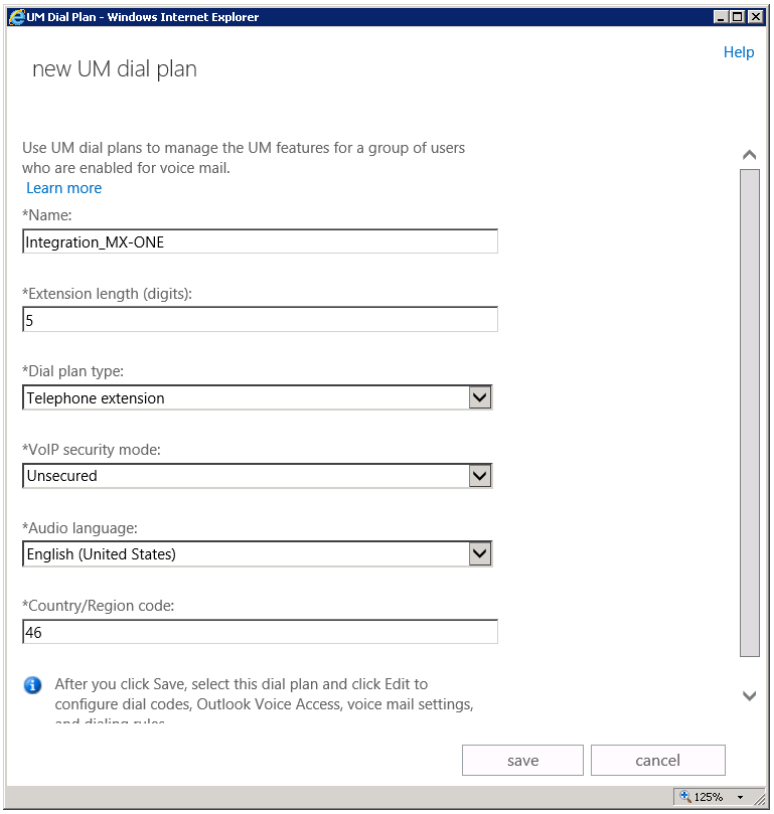
Before you create a UM dial plan, please read the Microsoft's document, UM Dial Plans.
http://technet.microsoft.com/en-us/library/bb125151(v=exchg.150).aspx

To create a New UM dial plan, please follow the step 1 in Microsoft's document, Create a UM dial plan.
http://technet.microsoft.com/en-us/library/jj673564(v=exchg.150).aspx

Example:

- UM Dial Plan: Integration_MX-ONE
- Number of digits in extensions numbers: 5 - It needs to match the number of digits of the MX-ONE extensions.
- VoIP Security: Unsecured. In this example TCP is used.

The screen below shows the required configuration for the example.

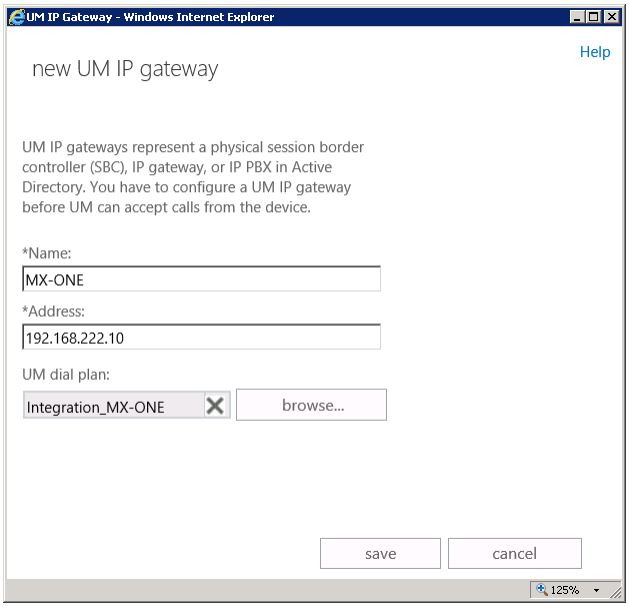


UM IP gateway

A UM IP gateway needs to be created in the Exchange UM.
To create a UM IP gateway, follow the step 2 in Microsoft's document:
http://technet.microsoft.com/en-us/library/jj673564(v=exchg.150).aspx

Example:

- Name of the gateway: MX-ONE
- IP address: 192.168.222.10
- Dial Plan: It is the same one created previously.

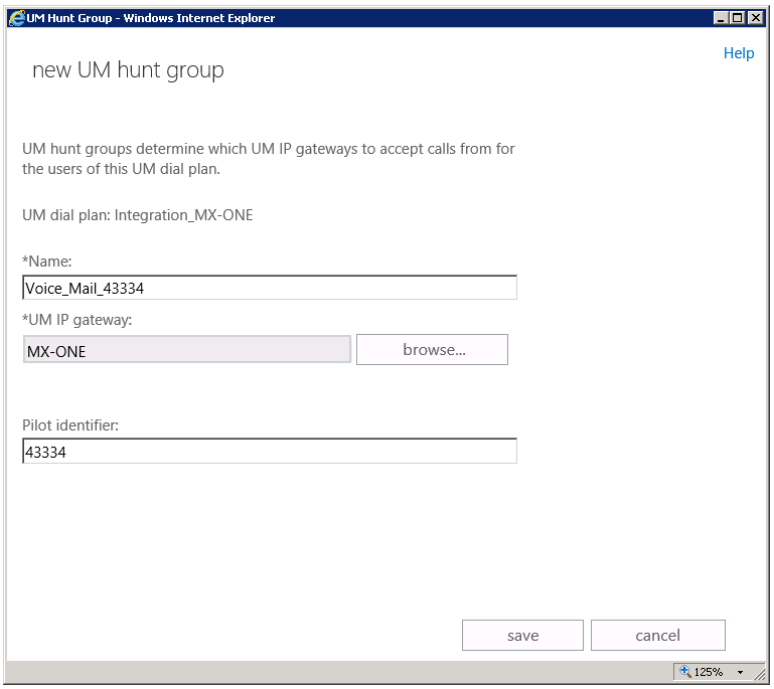

UM IP Gateway - Windows Internet Explorer
Help
new UM IP gateway
UM IP gateways represent a physical session border controller (SBC), IP gateway, or IP PBX in Active Directory. You have to configure a UM IP gateway before UM can accept calls from the device.
*Name:
MX-ONE
*Address:
192.168.222.10
UM dial plan:
Integration_MX-ONE
browse...
save
cancel
125%

UM Hunt Group

A Hunt group shall be created to the voice mail.
To create a UM Hunt Group, follow the steps in Microsoft's document:
http://technet.microsoft.com/en-us/library/aa997679.aspx.

Example:

- Associated UM IP Gateway: MX-ONE
- Name: Voice_Mail_43334
- Dial Plan: Integration_MX-ONE
- Pilot identifier: 43334. It must be the same number that was previously created in MX-ONE.



UM Mailbox Policies

A new UM mailbox policy can be created or the default policy can be used. Please, follow step 4 in Microsoft's document:
http://technet.microsoft.com/en-us/library/jj673564(v=exchg.150).aspx

UM Auto Attendant

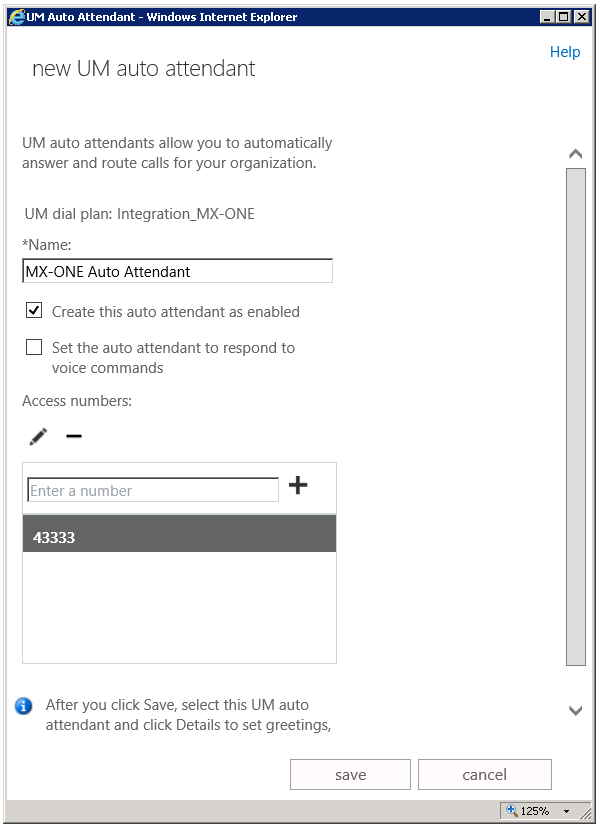
To setup the Exchange 2013 UM Auto Attendant, please follow the steps below:

Create an UM Auto Attendant

To create an UM Auto Attendant, please follow the step 5 in Microsoft's document:
http://technet.microsoft.com/en-us/library/jj673564(v=exchg.150).aspx

Example:

- Name: MX-ONE Auto Attendant
- Dial Plan: Integration_MX-ONE
- Pilot identifier: 43333. It needs to be the same number that was previously created in MX-ONE.


UM Auto Attendant - Windows Internet Explorer
Help
new UM auto attendant
UM auto attendants allow you to automatically answer and route calls for your organization.
UM dial plan: Integration_MX-ONE
*Name:
MX-ONE Auto Attendant
Create this auto attendant as enabled
Set the auto attendant to respond to voice commands
Access numbers:
Enter a number
43333
After you click Save, select this UM auto attendant and click Details to set greetings,
save
cancel
125%

Enable the Unified Messaging

To enable Voice Mail for a user follow Microsoft´s document:
Enable a User for Voice Mail
http://technet.microsoft.com/en-us/library/bb124147(v=exchg.150).aspx

5 MX-ONE Integration with Exchange 2013 UM Using TLS

5.1 Prerequisites

5.1.1 MX-ONE Prerequisites

Main components

Aastra MX-ONE Telephony System 5.0 SP4 or later with the proper licenses.
At least the following MX-ONE components are required:

- Aastra MX-ONE communications system
  - Telephony Server
    - MX-ONE Telephony Server 5.0 SP4 or later
  - Supported media gateways with the latest compatible firmware with MX-ONE 5.0 SP4 or later
    - MX-ONE Classic - 7U 19-inch chassis, using MGU boards or
    - MX-ONE Lite - 3U 19-inch chassis, using MGU board
    - MX-ONE Slim - 1U 19-inch chassis, using MGU board
  - Licenses
    - All licenses described in the item 3.1 MX-ONE Licenses
    - VoIP Encryption license is required (TLS/SRTP) as TLS and SRTP will be used.

The following shall be configured:

- Trunk between MX-ONE and Exchange UM – SIP route configured with TLS.

- Two IP extensions numbers to be used as Pilot numbers (groups) in Exchange UM.
- Message Waiting Indicator configuration in the system and in the phones that will use the service.
- Call list for IP phones. This feature is used to forward the call to the voice mail in case of no answer or busy.

5.1.2 Microsoft Exchange 2013 UM Prerequisites

This guide does not cover the Exchange 2013 UM installation. Our recommendation is that Microsoft Exchange 2013 UM shall be installed by a trained Microsoft engineer.
Before you start to install Microsoft Exchange 2013 Unified Messaging server role, please read the Microsoft Exchange 2013 documentation for a better understanding of the solution requirements, the documentation can be found in the following links:

- Microsoft Exchange 2013 documentation
  http://technet.microsoft.com/en-us/library/bb124558(v=exchg.150).aspx
- Microsoft Exchange 2013 Unified Messaging
  http://technet.microsoft.com/en-us/library/jj150478(v=exchg.150).aspx

5.2 Configuration

In this configuration example, we used the following:

Direct SIP connection using TLS as transport

MX-ONE Telephony System

- IP address: 192.168.222.10
- FQDN: mx-one-lync.lab.moon.galaxy
- Numbering Plan: 5 digits
- IP extensions numbers for Voice Mail and Auto Attendant: 43334 and 43333.
- Route access code: 043
- Users IP extensions: 27000, 27001 and 27010.

Exchange UM

- IPv4: 192.168.222.4
- FQDN: exc-2013-um1.moon.galaxy
- Voice Mail Pilot identifier, Hunt Group: 43334
- Auto Attendant: 43333

Certificate:
To use TLS between MX-ONE Telephony System and Exchange 2013 UM a certificate must be created.
The common Microsoft Enterprise CA used for signing server certificates for Mediation Server and Exchange 2013 UM is assumed to be used to create a server certificate for MX-ONE Telephony System as well.
A server certificate is signed to the FQDN (Fully Qualified Domain Name) of MX-ONE Telephony Server.

5.2.1 Create a Certificate

When using security, an appropriate certificate needs to be installed in MX-ONE as well as the encryption licenses. Please, check Certificate Management on MX-ONE CPI documentation in case you need more details regarding certificates.

5.2.2 Import the certificate to MX-ONE Telephony Server

Import the server certificate, in the example, mx-one-lync.lab.moon.galaxy.pfx to MX-ONE Telephony Server accessing the Exchange 2013 UM.
On the access Server, for example, Telephony Server 1 run the command:

Install certificate in MX-ONE Telephony Server
cert_install_local mx-one-lync.lab.moon.galaxy.pfx

5.2.3 MX-ONE configuration:

Creating SIP trunk with TLS

The following commands shall be executed in MX-ONE to configure a SIP Trunk with TLS, the others commands are the same as in a TCP configuration.

SIP Route settings for TLS
sip_route -set -route 55 -uristring sip:?@exc-2013-um1.moon.galaxy -fromuri0 sip:?@mx-one-lync.lab.moon.galaxy -remoteport 5061 -protocol tls -codecs PCMA -mwinumber 43334 # mwinumber is the Message Waiting Indication number

Inbound setting:
sip_route -set -route 55 -accept REMOTE_IP -match 192.168.222.4

Note that –accept REMOTE_IP will match the IP address send in the IPv4 source IP header.

Enable Media Encryption in the route:
media_encryption_enable -type route

5.2.4 Microsoft Exchange 2013 UM

In order to setup Exchange 2013 UM to use TLS, please follow Microsoft´s documentation.
http://technet.microsoft.com/en-us/library/jj150478(v=exchg.150).aspx

6 Load Balancing and Failover between MX-ONE and Two Exchange Servers

6.1 Load balancing

Aastra MX-ONE 5.0 SP4 supports load balancing when connected with more than one Exchange Server UM. To be able to use such a scenario, the Microsoft DNS Load Balancing functionality is used.
MX-ONE 5.0 and later supports DNS SRV and multiple A record query where a list with multiple entries can be used. When proper configured, MX-ONE will attempt to send INVITE to the entries in the list until the call is successful. No answer or 503 Service Unavailable will trigger MX-ONE to try the next entry.
For more details, check MX-ONE SIP Route command description in CPI or sip_route –help, parameter remoteport.

6.2 Failover

The failover functionality also requires Microsoft DNS Load Balancing functionality. When integrating MX-ONE and Exchange UM, the same configuration is valid for both failover and load balancing.
In a scenario where 2 Exchange UM servers are used and one of the servers is unavailable, the first call will be attempted to set up to the first server, but it will be redirected after a few seconds and answered. Then subsequent calls will be redirected and answered in the second Exchange UM.
The reason why it takes some seconds before getting answer is that the INVITE is sent to the first server, then the system waits 4 seconds for an answer. If no answer is received, the host is grey-listed for 32 seconds and an INVITE is sent to the second server.
For more details, check MX-ONE SIP Route command description in the CPI or sip_route –help, parameter remoteport.

6.3 Load balancing and Failover scenario

The figure below shows the validated setup:

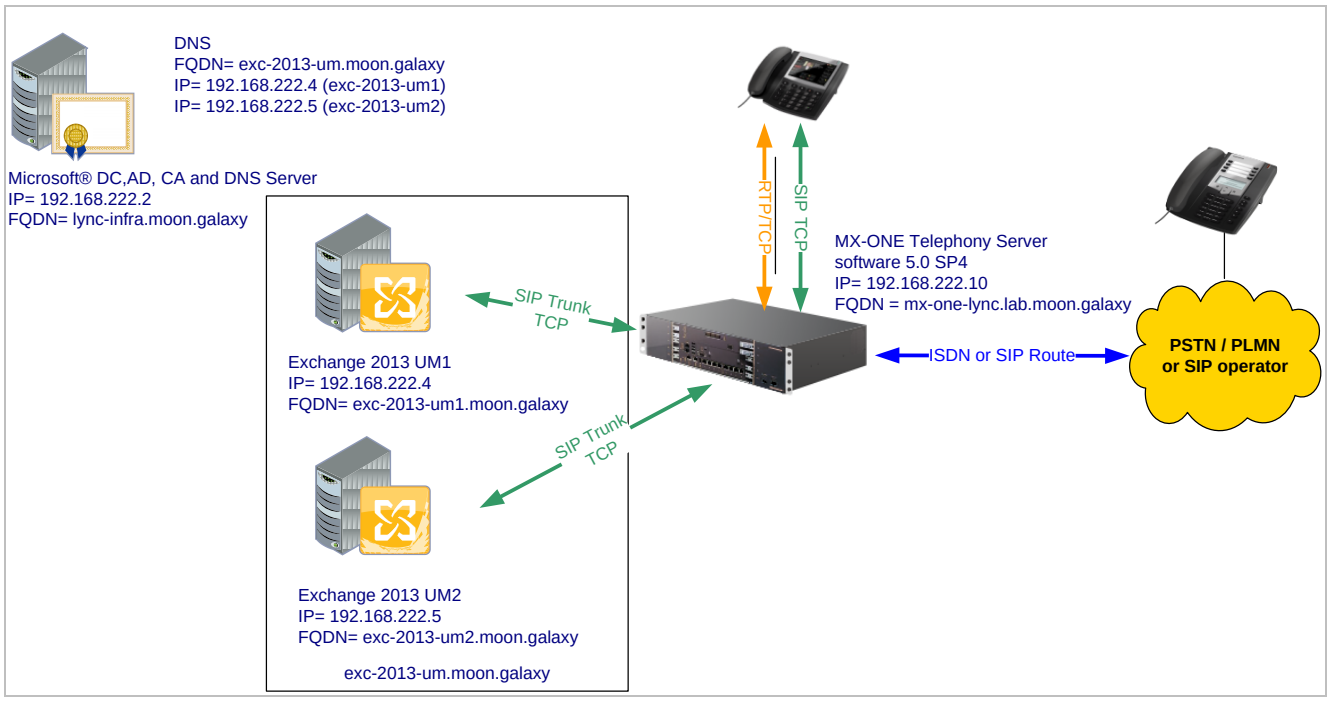


Figure 2 - Load Balancing and Failover setup

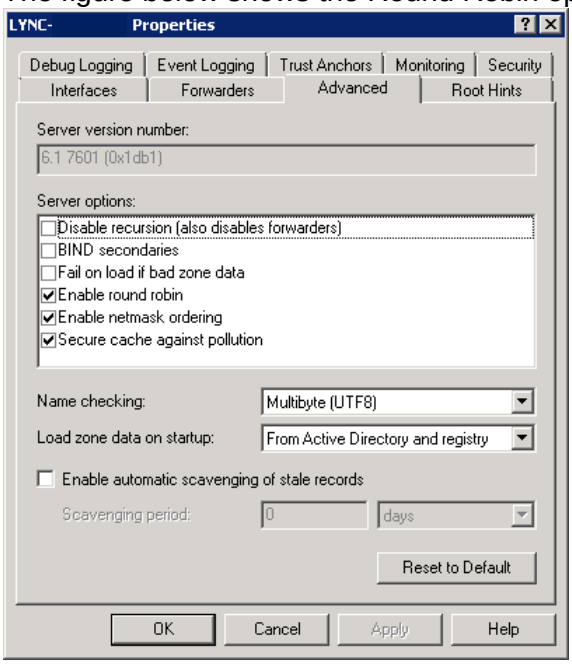
Configuration:

DNS Setup

Microsoft environment needs to be configured to support Round Robin as described in the TechNet article “Configure DNS for Load Balancing”.
Please, see the link below, item “To enable round robin for Windows Server”.

http://technet.microsoft.com/en-us/library/gg398251.aspx

The figure below shows the Round Robin option enabled.



DNS SRV setup

Go to DNS Manager Tool and create a pool entry. After that, add a DNS SRV record to each Exchange UM Server that participates in the DNS Load Balancing. In the following example the FQDN pool name is exc-2013-um.
The values below needs to be configured in the DNS SRV record for each Exchange UM that is part of the pool.

DNS SRV	Values
Service:	_sip
Protocol:	_tcp
Priority:	1
Weight:	1
Port Number:	5060
Host offering this service	exc-2013-um1 exc-2013-um2

Please check the following Microsoft article for more details.
http://technet.microsoft.com/en-us/library/gg398680

The figure below shows the exc-2013-um pool after the SRV Records configuration:

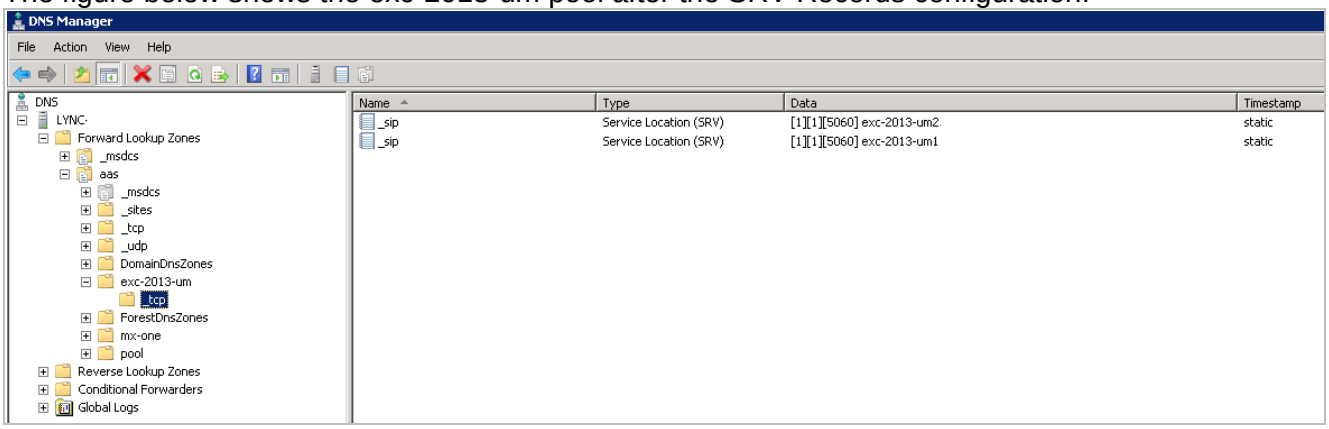


Figure 3 – DNS SRV records example

Test DNS SRV record setup

Using the Windows command nslookup, test the configuration:
nslookup
set type=srv
_sip._tcp.exc-2013-um

The expected result is presented in the 2 next screens. Please note the domain name and the IP addresses are just partially presented.

The first query the DNS replies with exc-2013-um1.

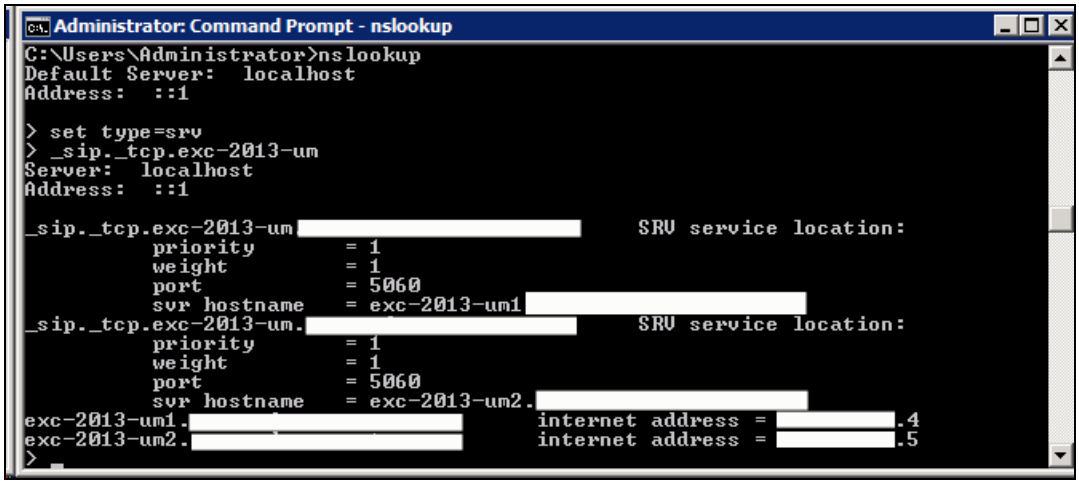


Figure 4 - DNS SRV example 1

The second query the DNS replies with exc-2013-um2.

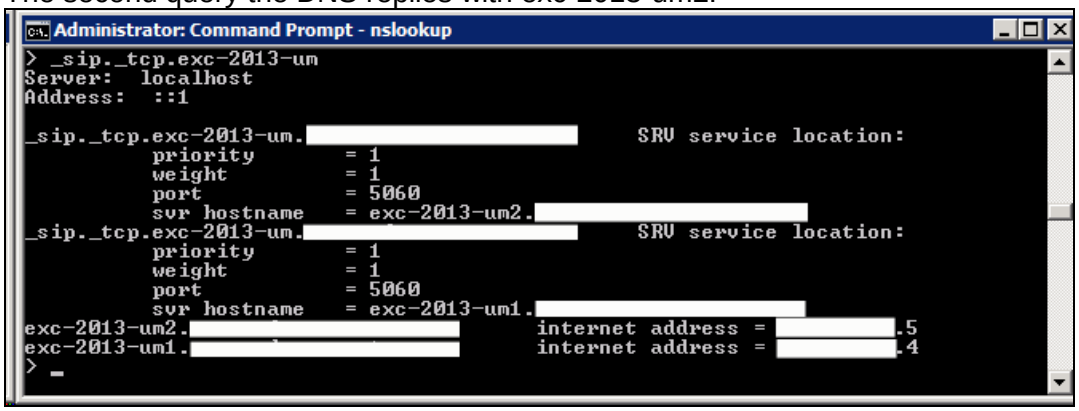


Figure 5 - DNS SRV example 1

Creating MX-ONE SIP trunk

The MX-ONE SIP Route Outbound setting needs to be configured with the FQDN pool name that is used to solve the Exchange UM Servers that are part of the “load balancing cluster”. Please, note that remoteport should be configured equal 0. This is needed by MX-ONE in order to use the DNS SRV option.

Outbound Setting:
sip_route -set -route 55 -uristring0 sip:?@exc-2013-um.moon.galaxy -remoteport 0 -protocol tcp -codecs PCMA,PCMU -mwinumber 43334 # mwinumber is the Message Waiting Indication number

Please note that Exchange 2013 UM IP addresses needs to be defined in the parameter **match**, as shown in the example.

Inbound Setting:
ip_route -set -route 55 -accept REMOTE_IP -match 192.168.222.4,192.168.222.5 # match = Exchange 2013 UM IP addresses

7 How to Test the Integration

To execute the integration test, the configuration in both sides shall be ready.

7.1 Basic Tests

1) Dial the pilot number from a phone extension that is NOT enabled for Unified Messaging and logon to a user's mailbox.
   Confirm hearing the prompt: "<Microsoft Exchange Earcon>. To access your mailbox, enter your extension…"
2) Navigate mailbox using the Voice User Interface (VUI).
3) Navigate mailbox using the Telephony User Interface (TUI).
4) Dial user extension and leave a voicemail.
   a) Dial user extension and leave a voicemail from an internal extension.
   Confirm that the Active Directory name of the calling party is displayed in the sender field of the voicemail message.
   b) Dial user extension and leave a voicemail from an external phone.
   Confirm that the correct phone number of the calling party is displayed in the sender field of the voicemail message.
5) Dial Auto Attendant (AA).
   Dial the extension for the AA and confirm that the AA answers the call.
6) Call Transfer by Directory Search.
   a) Call Transfer by Directory Search and have the called party answer.
   Confirm that the correct called party answers the phone.
   b) Call Transfer by Directory Search when the called party's phone is busy.
   Confirm that the call is routed to the called party's voicemail.
   c) Call Transfer by Directory Search when the called party does not answer.
   Confirm that the call is routed to the called party's voicemail.
   d) Setup an invalid extension number for a particular user.
   Call Transfer by Directory Search to this user. Confirm that the number is reported as invalid.
7) Outlook Web Access (OWA) Play-On-Phone Feature.
   a) Listen to voicemail using OWA's Play-On-Phone feature to a user's extension.
   Listen to voicemail using OWA's Play-On-Phone feature to an external number.
8) Configure a button on the phone of a UM-enabled user to forward the user to the pilot number.
   Press the voicemail button.
   Confirm that you are sent to the prompt: "<Microsoft Exchange UM Earcon>. <User>. Please enter your pin and press the pound key."
9) MWI.
   Ensure that a UM-enabled user's mailbox does not have any new voice mails.
   a) Dial the user's extension and leave a voicemail. Confirm that the MWI lamp on the phone lights up.
   b) Mark the voice mail email as read in OWA. Confirm that the MWI lamp on the phone turns off.
10) Load balancing
   a) Open the Wireshark tool and configure it to collect SIP packets.
   Dial several times to the voice mail number from an SIP extension. Use the Wireshark tool to analyze the SIP packet in order to verify that the load balancing is working properly.
11) Failover
   a) Disconnect the Ethernet cable of Exchange 2013 UM 1 to simulate a failure.
   Dial several times to the voice mail number from an SIP extension. Check that the calls are answered in the Exchange 2013 UM 2.

8 Revision History

Document Version	Comments	Date
Rev A	First version	January 2014

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